

BEST AVAILABLE COPY**PATENT ABSTRACTS OF JAPAN**

(11)Publication number : **10-207496**
 (43)Date of publication of application : **07.08.1998**

(51)Int.CI.

G10L 9/14
H03M 7/30
H04B 14/04

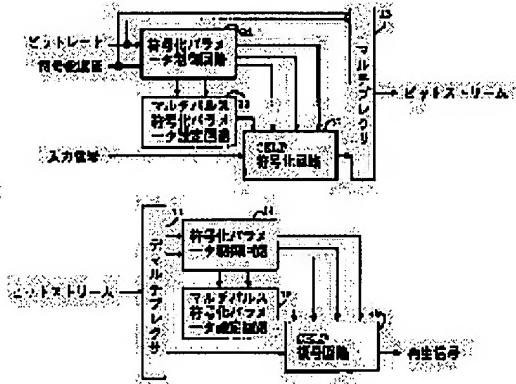
(21)Application number : **09-012477**
 (22)Date of filing : **27.01.1997**

(71)Applicant : **NEC CORP**
 (72)Inventor : **NOMURA TOSHIYUKI**

(54) VOICE ENCODING DEVICE AND VOICE DECODING DEVICE**(57)Abstract:**

PROBLEM TO BE SOLVED: To encode a voice signal in high quality by specified bit rate and an encoding delay.

SOLUTION: An encoding parameter control circuit 31 calculates a frame length from a bit rate and an encoding delay, and outputs it to a CELP encoding circuit 32. Based on the calculated frame length, the encoding parameter control circuit 31 selects in response to the bit rate a control parameter from a table in which two or more control parameters controlling an operation of CELP encoding circuit 32 are described, and outputs it to CELP encoding circuit 32. Further, the encoding parameter control circuit 31 outputs the number of bits distributed to a sub-frame length and a multiple pulse signal to a multiple pulse generation parameter setting circuit 33. The multiple pulse encoding parameter setting circuit 33 calculates the number of pulses representing a multi-pulse excitation signal from the sub-frame length and the number of bits of the multiple pulse signal, the number of pulse candidate positions of each pulse, and the number of pulse candidate positions.

**LEGAL STATUS**

[Date of request for examination]	27.01.1997
[Date of sending the examiner's decision of rejection]	19.12.2001
[Kind of final disposal of application other than the examiner's decision of rejection or application converted registration]	
[Date of final disposal for application]	
[Patent number]	3329216
[Date of registration]	19.07.2002
[Number of appeal against examiner's decision of rejection]	2002-01018
[Date of requesting appeal against examiner's decision of rejection]	18.01.2002
[Date of extinction of right]	

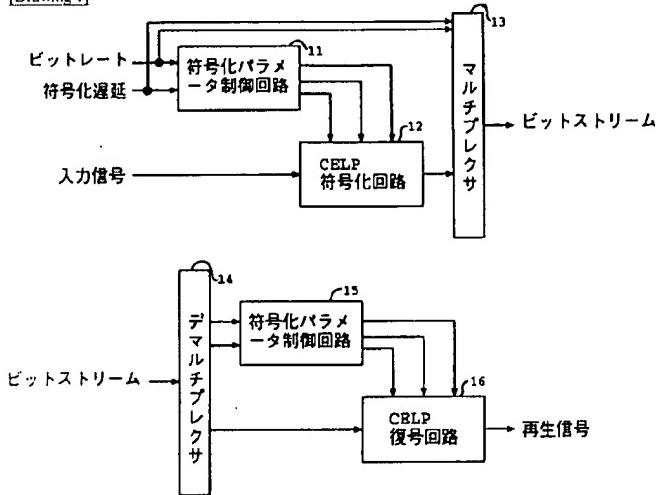
* NOTICES *

Japan Patent Office is not responsible for any damages caused by the use of this translation.

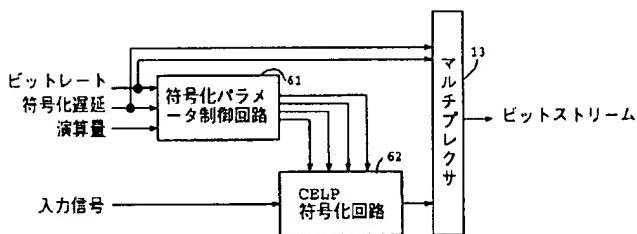
1. This document has been translated by computer. So the translation may not reflect the original precisely.
2. **** shows the word which can not be translated.
3. In the drawings, any words are not translated.

DRAWINGS

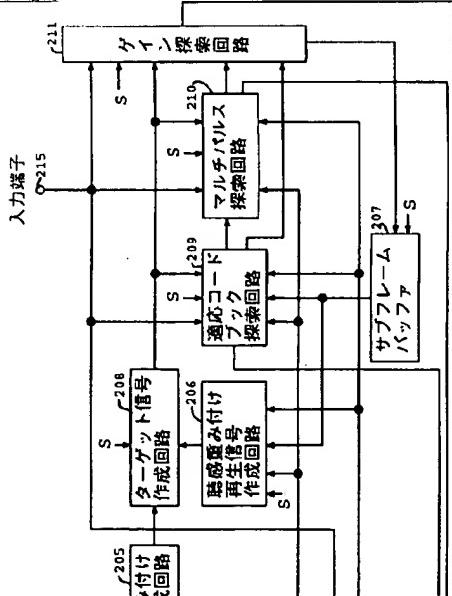
[Drawing 1]



[Drawing 7]

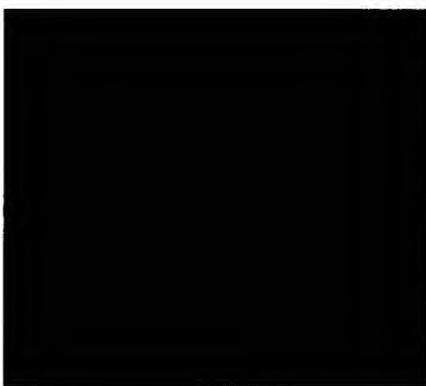
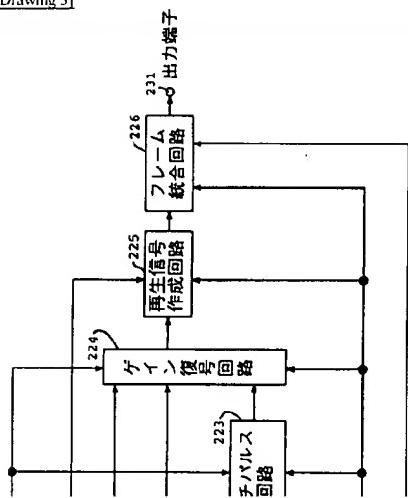


[Drawing 2]

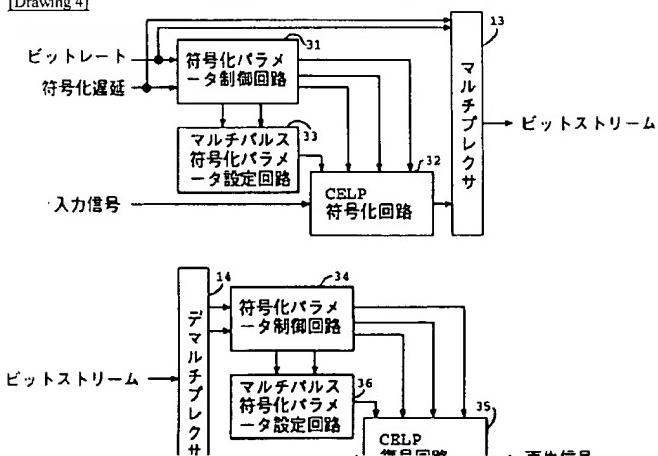




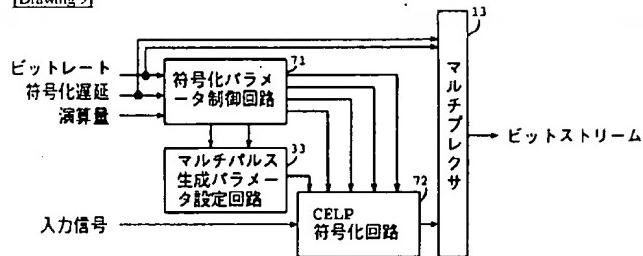
[Drawing 3]



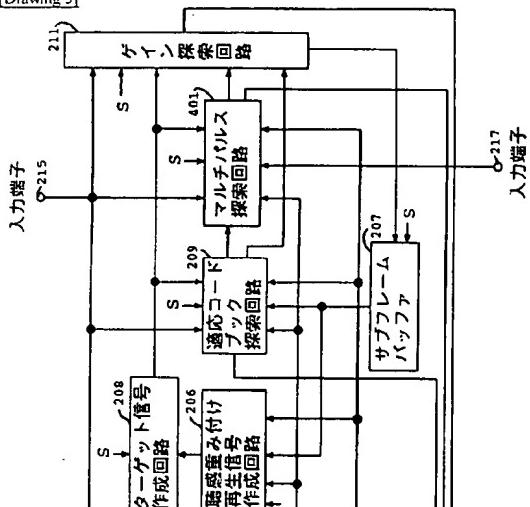
[Drawing 4]



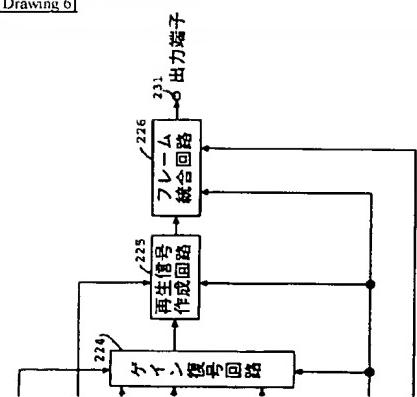
[Drawing 9]

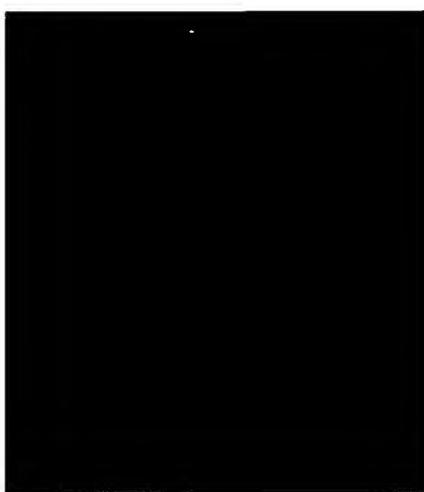


[Drawing 5]

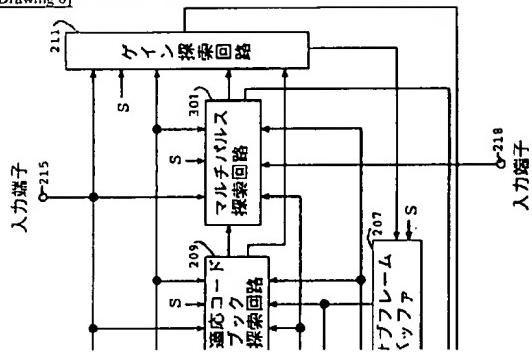


[Drawing 6]

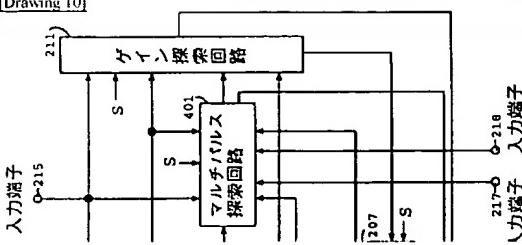


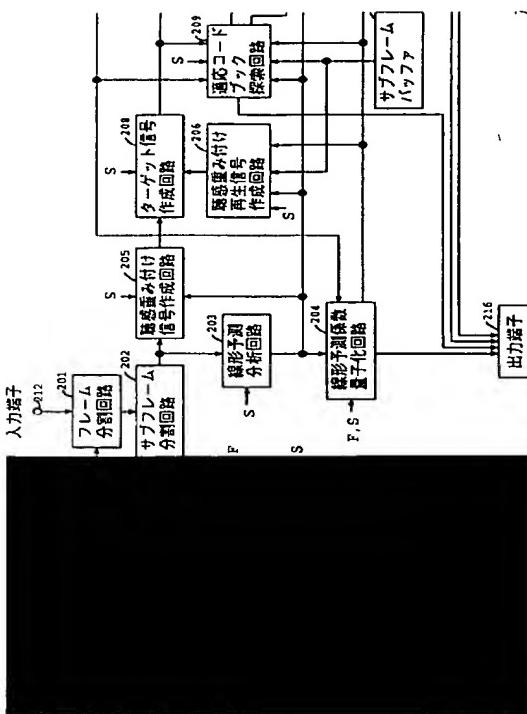


[Drawing 8]

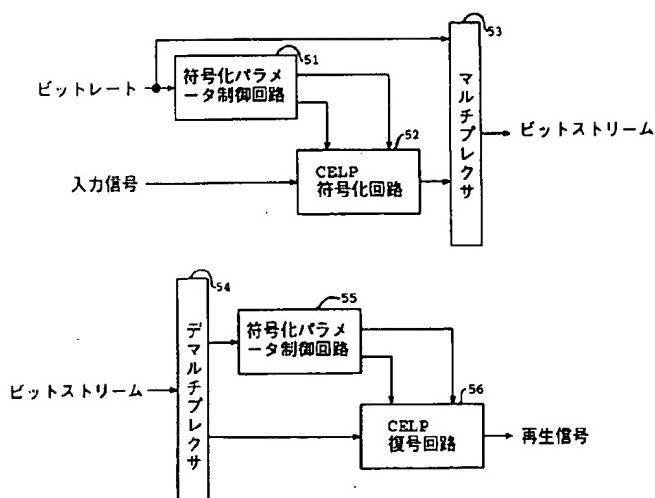


[Drawing 10]





[Drawing 11]



[Translation done.]

* NOTICES *

Japan Patent Office is not responsible for any damages caused by the use of this translation.

1. This document has been translated by computer. So the translation may not reflect the original precisely.
2. **** shows the word which can not be translated.
3. In the drawings, any words are not translated.

DETAILED DESCRIPTION

[Detailed Description of the Invention]

[0001]

[The technical field to which invention belongs] this invention relates to the voice to digital converter and voice decode equipment (it is called the voice coding decode equipment below) which are encoded with high quality with the parameter which specified the sound signal.

[0002]

[Description of the Prior Art] The equipment (method) indicated by the standardization advice specification (reference 1) generally entitled "Enhanced Variable Rate Codec, Speech Service Option 3 for Wideband Spread Spectrum Digital Systems" (information-separator-127, TIA.TR45) which are used for example, for the CDMA (Code Division Multiple Access) system as voice coding decode equipment which can control a bit rate is known.

[0003] By this method, a bit rate is determined according to the feature of an input signal, and the control parameter of a CELP (Code ExcitedLinear Prediction) coding method is set up from the table beforehand defined according to this determination result. And the input signal is encoded according to a setting control parameter. Furthermore, by this method, it also has the function to set up a bit rate compulsorily by the external signal.

[0004] Here, with reference to drawing 11, it outlines about this kind of voice coding decode equipment. According to an external signal, the bit rate is controlled by the voice coding decode equipment shown in drawing 11.

[0005] The voice coding decode equipment of illustration is equipped with a voice to digital converter and voice decode equipment, and a voice to digital converter and voice decode equipment are equipped with the coding parameter-control circuits 51 and 55, respectively. In a voice to digital converter, a bit rate is given to the coding parameter-control circuit 51, the control parameter which controls operation of the CELP coding network 52 chooses the control parameter corresponding to a bit rate with reference to the indicated table (ROM which does not illustrate but makes :, for example, a bit rate, the address (Read Only Memory)), and the coding parameter-control circuit 51 outputs this control parameter to the CELP coding network 52. A control parameter is the subframe length and bit distribution which are the batch of excitation signal coding in CELP coding.

[0006] The input signal (input sound signal) is given to the CELP coding network 52, and the linear predictor coefficients which were able to define the input signal beforehand and which carry out linear predictive coding for every frame, and express the spectral-envelope property of a sound signal are computed in the CELP coding network 52. And an excitation signal is computed by driving the linear-prediction composition filter corresponding to the spectral-envelope property, and an excitation signal is encoded according to bit distribution, respectively. And coding of an excitation signal divides a frame into a subframe further, and is performed for every subframe. This subframe length is set up by the coding parameter-control circuit 51 as mentioned above.

[0007] An above-mentioned excitation signal is constituted by the periodic component showing the pitch period of an input signal, the remaining remainder components, and those gain. The periodic component showing the pitch period of an input signal is expressed as an adaptation code vector stored in the code book holding the excitation signal of the past called adaptation code book. a remainder component For example Mr. J-P.Adoul -- ** -- it depends -- "Fast CELP coding based on algebraic It is expressed as a multi-pulse signal indicated by the paper (reference 2) it was [codes" (-1960 or Proc.ICASSP and pp.1957 1987)] entitled. And weighting addition of an adaptation code vector and the multi-pulse signal is carried out according to the gain held at the gain code book, and an excitation signal is generated. In addition, a regenerative signal is compoundable by driving a linear-prediction composition filter by the excitation signal.

[0008] Here, in case an adaptation code vector, a multi-pulse signal, and gain are chosen, a selection control is performed so that the error power after carrying out audibility weighting of the error signal between a regenerative signal and an input signal may serve as the minimum. And in the CELP coding network 52, an adaptation code vector, a multi-pulse signal and the index corresponding to gain, and the index corresponding to linear predictor coefficients are outputted to a multiplexer 53.

[0009] A multiplexer 53 changes and outputs an adaptation code vector, a multi-pulse signal and the index corresponding to gain, and the index corresponding to linear predictor coefficients to a bit stream for every frame. In addition, the information showing a bit rate is stored in a part for the header unit of a bit stream.

[0010] After extracting the information which expresses with voice decode equipment the bit rate which exists in a part for the header unit of a bit stream when a bit stream is received by the demultiplexer 54 and outputting to the coding

parameter-control circuit 55, an adaptation code vector, a multi-pulse signal and the index corresponding to gain, and the index corresponding to linear predictor coefficients are extracted from a bit stream for every frame, and it outputs to the CELP decoder circuit 56.

[0011] In the coding parameter-control circuit 55, the same processing as the coding parameter-control circuit 51 is performed, a control parameter is chosen according to the inputted bit rate information, and it outputs to the CELP decoder circuit 56.

[0012] In the CELP decoder circuit 56, while using an adaptation code vector, a multi-pulse signal and the index corresponding to gain, and the index corresponding to linear predictor coefficients, decode processing is performed using subframe length and a bit rate. An excitation signal carries out weighting addition of an adaptation code vector and the multi-pulse signal by the gain held at the gain code book, and is acquired by carrying out. and -- the CELP decoder circuit 56 -- an excitation signal -- a linear-prediction composition filter -- driving -- a regenerative signal -- ** -- it reproduces by carrying out

[0013] Thus, in a CELP coding method, in case a bit rate is controlled, the subframe length and bit distribution which are the batch of excitation signal coding are controlled, and a bit rate is controlled.

[0014]

[Problem(s) to be Solved by the Invention] By the way, there is a trouble that the coding delay defined by conventional voice coding decode equipment as time since the frame length which is a coding batch is a fixed value, after an input signal sample is inputted first until coding processing starts is uncontrollable.

[0015] Furthermore, it is necessary to have beforehand a parameter required in order to generate a multi-pulse signal, and, for this reason, there is a trouble that it can respond only to the bit rate set up beforehand, with conventional voice coding decode equipment.

[0016]

[Means for Solving the Problem] According to this invention The excitation signal of an input sound signal is expressed with the multi-pulse signal which consists of two or more pulses. with the aforementioned excitation signal It is the voice to digital converter which has a voice coding means to determine that the aforementioned excitation signal will minimize distortion between the reproduction sound signals and the aforementioned input sound signals which excite the linear-prediction composition filter specified by the linear predictor coefficients of the aforementioned input sound signal, and are obtained. The control circuit which receives the bit rate and coding delay which were specified as control information, and generates a control parameter according to this control information, It has the setting circuit which sets up a parameter required for coding of the aforementioned multi-pulse signal as an active parameter according to a **** parameter. the parameter beforehand defined among the aforementioned control parameters gives -- having -- eye this ** -- laws -- The voice to digital converter characterized by the aforementioned voice coding means encoding the aforementioned input sound signal based on the aforementioned control parameter and the aforementioned active parameter is obtained.

[0017] Furthermore, according to this invention, it is voice decode equipment which receives coding voice data and reproduces a reproduction sound signal from the aforementioned coding voice data. The aforementioned coding voice data includes a bit rate and coding delay while having the excitation signal and linear-prediction composition filter factor of a sound signal. The control circuit which the aforementioned excitation signal is expressed with the multi-pulse signal which consists of two or more pulses, and generates a control parameter based on the aforementioned bit rate and the aforementioned coding delay, the parameter beforehand defined among the aforementioned control parameters gives -- having -- eye this ** -- laws -- with the setting circuit which sets up a parameter required for coding of the aforementioned multi-pulse signal as an active parameter according to a **** parameter Based on the aforementioned control parameter and the aforementioned active parameter, the aforementioned excitation signal and the aforementioned linear-prediction composition filter factor are decoded from the aforementioned coding voice data. The voice decode equipment characterized by having a decode means to excite the linear-prediction composition filter specified by the aforementioned linear-prediction composition filter factor according to the aforementioned excitation signal, and to reproduce the aforementioned reproduction sound signal is obtained.

[0018] That is, at the voice coding network by this invention, it has the coding parameter-control circuit which generates frame length required in order to encode by the bit rate and coding delay which were specified, subframe length, and bit distribution as a control parameter, and an input sound signal is divided into a frame by the set-up frame length. And in a multi-pulse generation parameter setup circuit, a parameter required in order to generate a multi-pulse signal from the specified bit rate and coding delay is set up.

[0019] Since frame length, subframe length, and bit distribution are generated in a coding parameter-control circuit and the input sound signal is divided into the frame according to this frame length, the frame length which is a coding batch is made to adjustable. For this reason, in addition to a bit rate, coding delay can also be controlled.

[0020] Furthermore, in a multi-pulse generation parameter setup circuit, the flexibility of the bit rate which can be specified increases by setting up the parameter for generating a multi-pulse signal. For this reason, it is not necessary to set up beforehand the bit rate which can respond.

[0021]

[Embodiments of the Invention] this invention is explained with reference to a drawing below.

[0022] With reference to drawing 1, the voice coding decode equipment of illustration has a voice to digital converter and voice decode equipment, and the voice to digital converter is equipped with the coding parameter-control circuit 11, the

CELP coding network 12, and the multiplexer 13. On the other hand, voice decode equipment is equipped with the demultiplexer 14, the coding parameter-control circuit 15, and the CELP decoder circuit 16.

[0023] In a voice to digital converter, frame length is computed by a bit rate and coding delay being given to the coding parameter-control circuit 11 as control information, and lengthening read-ahead length required for the analysis processing in CELP coding from these bit rates and coding delay. For example, frame length is set to 20ms when coding delay is [the read-ahead length of linear predictive coding] 5ms in 25ms.

[0024] Furthermore, in the coding parameter-control circuit 11, the control parameter which controls operation of the CELP coding network 12 based on the frame length which computed chooses a control parameter according to an input bit rate from the indicated tables, and outputs to the CELP coding network 12. Control parameters are frame length, subframe length (for example, 5ms), and bit distribution. To the CELP coding network 12, an input signal (input sound signal) is given and encodes an input signal according to the set-up frame length, subframe length, and bit distribution.

[0025] Here, drawing 2 is also referred to and operation of the CELP coding network 12 is explained.

[0026] The frame length (F) set up by the coding parameter-control circuit 11 is inputted into the frame dividing network 201 and the linear-predictor-coefficients quantization circuit 204 through an input terminal 213.

[0027] On the other hand, it is set up by the coding parameter-control circuit 11, and subframe length (S) is inputted into the subframe dividing network 202, the linear-predictive-coding circuit 203, the linear-predictor-coefficients quantization circuit 204, the audibility weighting signal creation circuit 205, the audibility weighting regenerative-signal creation circuit 206, the target signal creation circuit 208, the adaptation code book search circuit 209, the multi-pulse search circuit 210, and the gain search circuit 211 through an input terminal 214.

[0028] Furthermore, bit distribution in each parameter set up by the coding parameter-control circuit 11 is inputted into the linear-predictor-coefficients quantization circuit 204, the adaptation code book search circuit 209, the multi-pulse search circuit 210, and the gain search circuit 211 through an input terminal 215.

[0029] According to setting frame length (F), the frame dividing network 201 divides for every frame, is got blocked and outputs an input signal to the subframe dividing network 202 for every frame.

[0030] The subframe dividing network 202 divides a frame further according to setting subframe length (S), considers as a subframe, and outputs this subframe to the linear-predictive-coding circuit 203 and the audibility weighting signal creation circuit 205.

[0031] The linear-predictive-coding circuit 203 carries out linear predictive coding of the signal (subframe signal) given from the subframe dividing network 202 for every subframe based on setting subframe length (S), and outputs linear-predictor-coefficients $a'(i)$, $i=1, \dots, N_p$ to the linear-predictor-coefficients quantization circuit 204, the audibility weighting signal creation circuit 205, the audibility weighting regenerative-signal creation circuit 206, the adaptation code book search circuit 209, and the multi-pulse search circuit 210. Here, N_p is the degree of linear predictive coding, for example, is 10. It is explained in full detail by the reference (Tokai University Press) (reference 3) which there are a correlation method and a covariance method as linear-predictive-coding method, for example, was entitled the "digital speech processing" by Furui.

[0032] In the linear-predictor-coefficients quantization circuit 204, the linear predictor coefficients obtained for every subframe are put in block with a frame according to setting frame length (F) and setting subframe length (S), and it quantizes. Under the present circumstances, in order to reduce a bit rate, it quantizes by the subframe of the last in a frame, and the technique for which the quantization value of other subframes uses the interpolation value of the quantization value of the frame concerned and the last frame is used. And this quantization and interpolation are performed after changing linear predictor coefficients into a line spectrum pair (henceforth referred to as loop splice plate). In addition, the conversion to loop splice plate from linear predictor coefficients is indicated by the paper (-606 or Institute of Electronics and Communication Engineers paper magazine, J64-A, and pp.599 1981) (reference 4) entitled "speech information compression by the line-spectrum-pair (loop splice plate) voice-analysis composite system". [else / Sugamura] Moreover, the well-known technique can be used for the method of quantizing loop splice plate. For example, about the method of quantizing loop splice plate, since it is indicated by JP,4-171500,A (reference 5), explanation is omitted here. In the linear-predictor-coefficients quantization circuit 204, Quantization loop splice plate is changed into linear predictor coefficients, and it outputs to the audibility weighting signal creation circuit 205, the audibility weighting regenerative-signal creation circuit 206, the adaptation code book search circuit 209, and the multi-pulse search circuit 210 as quantization linear-predictor-coefficients $a'(i)$, $i=1, \dots, N_p$.

[0033] And the index showing Quantization loop splice plate is outputted to a multiplexer 13 through an output terminal 216. In addition, the linear-prediction composition filter $H_s(z)$ is expressed with several 1.

[0034]

[Equation 1]

$$H_s(z) = \frac{1}{1 - \sum_{i=1}^{N_p} a'(i) z^{-i}}$$

In the audibility weighting signal creation circuit 205, the audibility weighting filter $H_w(z)$ expressed with several 2 is constituted using linear predictor coefficients, an audibility weighting filter is driven by the input signal in a subframe, and an

audibility weighting signal is created. And this audibility weighting signal is outputted to the target signal creation circuit 208.

[0035]

[Equation 2]

$$H_w(z) = \frac{1 - \sum_{i=1}^{N_p} a(i) R_2^{-1} z^{-i}}{1 - \sum_{i=1}^{N_p} a(i) R_1^{-1} z^{-i}}$$

Here, R1 and R2 are weighting factors which control the amount of audibility weighting. For example, it is R1=0.6 and R2=0.9.

[0036] In the audibility weighting regenerative-signal creation circuit 206, the linear-prediction composition filter of the subframe before [one] holding in this circuit using the excitation signal of the subframe before [one] being obtained through the subframe buffer 207, and an audibility weighting composition filter are driven, and the state of both the filters after a drive is outputted to the target signal creation circuit 208.

[0037] In the target signal creation circuit 208, after inputting the state of the linear-prediction composition filter obtained from the audibility weighting regenerative-signal creation circuit 206, and an audibility weighting filter, creating the zero input response of the filter which made continuation connection of both the filters and subtracting from an audibility weighting signal, it outputs to the adaptation code book search circuit 209, the multi-pulse search circuit 210, and the gain search circuit 211 as a target signal.

[0038] In the adaptation code book search circuit 209, after updating the code book holding the excitation signal of the past called adaptation code book by the excitation signal of the subframe before [one] being obtained through the subframe buffer 207, the adaptation code vector corresponding to Pitch d is chosen from an adaptation code book. Here, it connects repeatedly and an adaptation code vector is created until Pitch d is late for the excitation signal of the past stored in the adaptation code book, starts the segment of d and becomes subframe length, in being shorter than subframe length. And using the created adaptation code-vector signal Ad (n), a linear-prediction composition filter and an audibility weighting filter are driven in the state of zero, a regenerative signal SAd (n) is created, and the pitch d which makes the minimum the error Ed of the target signal X (n) and regenerative signal SAd (n) which are expressed with several 3 is chosen.

[0039]

[Equation 3]

$$E_d = \sum_{n=1}^L X(n)^2 - \frac{\left(\sum_{n=1}^L X(n) SAd(n) \right)^2}{\sum_{n=1}^L SAd(n)^2}$$

Here, L is the subframe length set up by the coding parameter-control circuit 11. Furthermore, the adaptation code book search circuit 209 outputs the adaptation code-vector signal Ad (n) chosen while outputting the selected pitch d to the multiplexer 13 through the output terminal 216, and its regenerative signal SAd (n) to the gain search circuit 211. Moreover, the adaptation code book search circuit 209 outputs a regenerative signal SAd (n) to the multi-pulse search circuit 210 while outputting a regenerative signal SAd (n) to the gain search circuit 211.

[0040] In the multi-pulse search circuit 210, a multi-pulse signal consists of pulses of two or more non-zero. Here, each pulse is chosen from the pulse-position candidate beforehand set for every pulse. Each pulse amplitude is only polarity. For example, when subframe length is 5ms in a 8kHz sampling (N= 40 sample), a multi-pulse excitation signal consists of pulses of P (for example, 5) individual. P pulses are chosen from the pulse candidate position of M (p) defined beforehand, respectively, p= 0, --, P-1 (each 8 [for example,]) individual. The multi-pulse search circuit 210 is carrying out ***** maintenance of what combined pulse-number P and the pulse candidate position of M (p) individual of each pulse, and chooses the combination of pulse-number P and the pulse candidate position of M (p) individual of each pulse according to the bit distribution specified by the coding parameter-control circuit 11. Using the selected pulse candidate position of M pieces of pulse-number P (equal to the number of channels), and each channel, the multi-pulse signal Cj (n) is created and the multi-pulse signal Cj (n) which minimizes several 4 is chosen.

[0041]

[Equation 4]

$$E_j = \sum_{n=1}^L X'(n)^2 - \frac{\left(\sum_{n=1}^L X'(n) SCj(n) \right)^2}{\sum_{n=1}^L SCj(n)^2}$$

Here, $X'(n)$ is the signal which subtracted the regenerative signal SAd of an adaptation code vector (n) from the target signal $X(n)$, and is given by several 5.

[0042]

[Equation 5]

$$X'(n) = X(n) - \frac{\sum_{n=1}^L X(n) SAd(n)}{\sum_{n=1}^L SAd^2(n)}$$

In addition, in case several 4 is minimized, the amount of operations is reduced using the technique indicated by for example, the Japanese-Patent-Application-No. No. 318071 [seven to] specification (reference 6). Furthermore, the multi-pulse search circuit 210 outputs the index j which corresponds while outputting the selected multi-pulse signal $C_j(n)$ and its regenerative signal $SC_j(n)$ to the gain search circuit 211 to a multiplexer 13 through an output terminal 216.

[0043] Furthermore, in the gain search circuit 211, using the regenerative signal SAd of an adaptation code vector (n), the regenerative signal SC_j of a multi-pulse (n), and the target signal $X(n)$, Gain GA and GC is quantized so that several 6 may be minimized.

[0044]

[Equation 6]

$$E_k = \sum_{n=1}^L (X(n) - G_k(1) SAd(n) - G_k(2) SC_j(n))^2$$

Moreover, in the gain search circuit 211, an excitation signal is created using the quantized gain, an adaptation code vector, and a multi-pulse signal, an excitation signal is outputted to the audibility weighting regenerative-signal creation circuit 206 and the adaptation code book search circuit 209 through the subframe buffer 207, and the index k corresponding to gain is outputted to a multiplexer 13 through an output terminal 216.

[0045] Again, with reference to drawing 1, the index which expresses Quantization loop splice plate with a multiplexer 13, the index of a pitch and a multi-pulse signal, and the index showing quantization gain are changed into a bit stream for every frame, and it outputs. In addition, the information showing a bit rate and coding delay is stored in a part for the header unit of a bit stream.

[0046] In voice decode equipment, a bit stream is given to a demultiplexer 14, and after outputting the information which expresses with a demultiplexer 14 the bit rate which exists in a part for the header unit of a bit stream, and coding delay to the coding parameter-control circuit 15, the index showing Quantization loop splice plate, the index of a pitch and a multi-pulse signal, and the index showing quantization gain are extracted from a bit stream for every frame, and it outputs to the CELP decoder circuit 16.

[0047] The coding parameter-control circuit 15 performs the same operation as the coding parameter-control circuit 11 by the side of coding, chooses a control parameter according to the bit rate and coding delay which were inputted, and outputs it to the CELP decoder circuit 16.

[0048] Drawing 3 is also referred to and operation of the CELP decoder circuit 16 is explained.

[0049] The index showing Quantization loop splice plate, the index of a pitch and a multi-pulse signal, and the index showing quantization gain are inputted into the linear-predictor-coefficients decoder circuit 221, the adaptation code book decoder circuit 222, the multi-pulse decoder circuit 223, and the gain decoder circuit 224 through an input terminal 227.

[0050] The frame length set up by the coding parameter-control circuit 15 is inputted into the linear-predictor-coefficients decoder circuit 221 and the frame integrated circuit 226 through an input terminal 228.

[0051] The subframe length set up by the coding parameter-control circuit 15 is inputted into the linear-predictor-coefficients decoder circuit 221, the adaptation code book decoder circuit 222, the multi-pulse decoder circuit 223, the gain decoder circuit 224, the regenerative-signal creation circuit 225, and the frame integrated circuit 226 through an input terminal 229.

[0052] The bit distribution set up by the coding parameter-control circuit 15 is inputted into the linear-predictor-coefficients decoder circuit 221, the adaptation code book decoder circuit 222, the multi-pulse decoder circuit 223, and the gain decoder circuit 224 through an input terminal 230.

[0053] In the linear-predictor-coefficients decoder circuit 221, the index showing Quantization loop splice plate is inputted for every frame, quantization linear-predictor-coefficients $a'(i)$, $i = 1, \dots, N_p$ are decoded for every subframe, and it outputs to the regenerative-signal composition circuit 225.

[0054] In the adaptation code book decoder circuit 222, an adaptation code vector is decoded from the pitch inputted for every subframe, and it outputs to the gain decoder circuit 224. In the multi-pulse decoder circuit 223, a multi-pulse signal is decoded from the index inputted for every subframe, and it outputs to the gain decoder circuit 224.

[0055] In the gain decoder circuit 224, gain is decoded from the index inputted for every subframe, an excitation signal is created using an adaptation code vector, a multi-pulse signal, and gain, and it outputs to the regenerative-signal composition circuit 225.

[0056] In the regenerative-signal composition circuit 225, the linear-prediction composition filter $H_s(z)$ is driven by the excitation signal for every subframe, a regenerative signal is created, and it outputs to the frame integrated circuit 226. In

addition, the linear-prediction composition filter $H_s(z)$ is expressed with the above-mentioned several 1. The regenerative signal inputted for every subframe is connected by frame length, and the frame integrated circuit 226 outputs it for every frame.

[0057] With reference to drawing 4, other examples of the voice coding decode equipment by this invention are explained.

[0058] The voice coding decode equipment of illustration has a voice to digital converter and voice decode equipment, and the voice to digital converter is equipped with the coding parameter-control circuit 31, the CELP coding network 32, the multi-pulse-coding parameter setup circuit 33, and the multiplexer 13. On the other hand, voice decode equipment is equipped with the demultiplexer 14, the coding parameter-control circuit 34, the CELP decoder circuit 35, and the multi-pulse-coding parameter setup circuit 16.

[0059] In a voice to digital converter, it computes frame length by the coding parameter-control circuit 31 receiving a bit rate and coding delay as control information, and lengthening read-ahead length required for the analysis processing in CELP coding from the bit rate and coding delay which were inputted. Moreover, the control parameter which controls operation of the CELP coding network 32 based on the frame length which computed chooses a control parameter according to the bit rate inputted out of the indicated table, and outputs to the CELP coding network 32. Control parameters are frame length, subframe length, and bit distribution. Furthermore, the coding parameter-control circuit 31 outputs the number of bits distributed to subframe length and the multi-pulse signal to the multi-pulse generation parameter setup circuit 33.

[0060] In the multi-pulse generation parameter setup circuit 33, from the subframe length N which inputted, and number-of-bits [of a multi-pulse signal] Y , pulse-number P required for coding of a multi-pulse excitation signal, number of each pulse] of pulse candidate positions $M(p)$, and its candidate position are computed so that several 7 and several 8 may be filled. Here, the pulse candidate position of each pulse is set up in sequence of numbers 0, 2, and 3, --, the form where $N-1$ was interleaved by pulse-number P as indicated by the above-mentioned reference 2. For example, when subframe length is [the number of bits of 40 samples ($N=40$) and a multi-pulse signal] 20 bits ($Y=20$), 5 and number of pulse candidate positions $M(p)$ is set to 8 by pulse-number P . The example of the pulse candidate position in this case is shown in Table 1.

[0061]

[Equation 7]

$$Y = \sum_{p=0}^{P-1} (1 + \log_2 M(p))$$

[0062]

[Equation 8]

$$N = \sum_{p=0}^{P-1} M(p)$$

[0063]

[Table 1]

表1：パルス候補位置の例

パルス番号	パルス候補位置
0	0, 5, 10, 15, 20, 25, 30, 35
1	1, 6, 11, 16, 21, 26, 31, 36
2	2, 7, 12, 17, 22, 27, 32, 37
3	3, 8, 13, 18, 23, 28, 33, 38
4	4, 9, 14, 19, 24, 29, 34, 39

The CELP coding network 32 encodes an input signal based on the frame length set up by the coding parameter-control circuit 31, subframe length and bit distribution, and pulse-number P set up in the multi-pulse generation parameter setup circuit 33, number [of each pulse] of pulse candidate positions $M(p)$ and its candidate position.

[0064] Drawing 5 is also referred to and operation of the CELP coding network 32 is explained.

[0065] Only operation of a multi-pulse search circuit differs compared with the CELP coding network which explained this CELP coding network 32 by drawing 2. Therefore, only operation of the multi-pulse search circuit 401 is explained here.

□AWINGS□ON OF DRAWINGS□earch circuit 401 inputs pulse-number P and the pulse candidate position of $M(p)$ individual pulse which were set up through the input terminal 217 in the multi-pulse generation parameter setup circuit 33, creates the multi-pulse signal $C_j(n)$, and chooses the multi-pulse signal $C_j(n)$ which minimizes the above-mentioned several 4. In addition, in the case of minimization of several four, the amount of operations can be reduced by using the technique of a publication for reference 6 as mentioned above.

[0067] Moreover, the multi-pulse search circuit 401 outputs the index j which corresponds while outputting the selected multi-pulse signal $C_j(n)$ and its regenerative signal $SC_j(n)$ to the gain search circuit 211 to a multiplexer 13 through an output terminal 216. And as it explained in relation to drawing 1, a multiplexer 13 outputs a bit stream.

[0068] With reference to drawing 4, a bit stream is received by the demultiplexer 14 in voice decode equipment. And as explained in relation to drawing 1, after a demultiplexer 14 outputs the information showing the bit rate which exists in a part for the header unit of a bit stream, and coding delay to the coding parameter-control circuit 34, it extracts the index showing Quantization loop splice plate, the index of a pitch and a multi-pulse signal, and the index showing quantization gain from a bit stream for every frame, and outputs them to the CELP decoder circuit 35.

[0069] The coding parameter setup circuit 34 performs the same operation as the coding parameter-control circuit 31, chooses a control parameter, and outputs it to the CELP decoder circuit 35.

[0070] The multi-pulse generation parameter setup circuit 36 computes the pulse number showing a multi-pulse excitation signal, the number of pulse candidate positions of each pulse, and its candidate position by performing the same operation as the multi-pulse generation parameter setup circuit 33 by the side of coding, and outputs them to the CELP decoder circuit 35.

[0071] Drawing 6 is also referred to and operation of the CELP decoder circuit 35 is explained.

[0072] Only operation of a multi-pulse decoder circuit differs compared with the CELP decoder circuit which explained this CELP decoder circuit 35 by drawing 3. Therefore, only operation of the multi-pulse decoder circuit 402 is explained here.

[0073] In the multi-pulse decoder circuit 402, a multi-pulse signal is decoded from the index which inputted the subframe length set up by the coding parameter-control circuit 34 through the input terminal 229, inputted the pulse number set up in the multi-pulse generation parameter setup circuit 36, the number of pulse candidate positions of each pulse, and its candidate position through the input terminal 232, and was inputted for every subframe.

[0074] With reference to drawing 7, the example of further others of the voice to digital converter by this invention is explained.

[0075] The voice to digital converter of illustration is equipped with the coding parameter-control circuit 61, the CELP coding network 62, and the multiplexer 13. The coding parameter-control circuit 61 performs the same operation as the coding parameter-control circuit 11 explained by drawing 1, and sets up frame length, subframe length, and bit distribution from the inputted bit rate and coding delay. Furthermore, in the coding parameter-control circuit 61, the amount of multi-pulse-coding permissible operations which is the amount of operations which can be spent on coding of a multi-pulse signal is computed from the inputted amount of operations. This memorizes the amount of operations required for coding of other parameters beforehand, and can compute it by deducting these values from the inputted amount of operations. The coding parameter-control circuit 61 is outputted to the CELP coding network 62 by using frame length, subframe length, and bit distribution and the amount of multi-pulse-coding permissible operations as a control parameter.

[0076] The CELP coding network 62 encodes an input signal according to the above-mentioned frame length, subframe length, and bit distribution and the amount of multi-pulse-coding permissible operations.

[0077] Drawing 8 is also referred to and operation of the CELP coding network 62 is explained.

[0078] Only operation of a multi-pulse search circuit differs compared with the CELP coding equipment which explained this CELP coding network 62 by drawing 2. Therefore, only the multi-pulse search circuit 301 is explained here.

[0079] The multi-pulse search circuit 301 performs the same operation as the multi-pulse search circuit 210 explained by drawing 2, and chooses the multi-pulse signal $C_j(n)$ which minimizes the above-mentioned several 4. In this case, preliminary selection is performed so that the amount of operations spent on coding of a multi-pulse signal may not exceed the amount of multi-pulse-coding permissible operations inputted through the input terminal 218. This preliminary selection is realizable by choosing what has the large value of E_1 expressed with several 9.

[0080]

[Equation 9]

$$E_1 = \left(\sum_{n=1}^L X'(n) S C_j(n) \right)^2$$

Moreover, the multi-pulse search circuit 301 outputs the index j which corresponds while outputting the selected multi-pulse signal $C_j(n)$ and its regenerative signal $S C_j(n)$ to the gain search circuit 211 to a multiplexer 13 through an output terminal 216.

[0081] With reference to drawing 9, the example of further others of the voice to digital converter by this invention is explained.

[0082] The voice to digital converter of illustration is equipped with the coding parameter-control circuit 71, the multi-pulse generation parameter setup circuit 33, the CELP coding network 72, and the multiplexer 13.

[0083] The coding parameter-control circuit 71 performs the same operation as the coding parameter-control circuit 31 explained by drawing 4, and sets up frame length, subframe length, and bit distribution from the bit rate and coding delay which were inputted. Furthermore, the coding parameter-control circuit 71 computes the amount of multi-pulse-coding permissible operations which is the amount of operations which can be spent on coding of a multi-pulse signal from the inputted amount of operations. And the coding parameter-control circuit 71 outputs frame length, subframe length, and bit distribution and the amount of multi-pulse-coding permissible operations to the CELP coding network 72. Furthermore, the coding parameter-control circuit 71 outputs the number of bits distributed to the multi-pulse signal with subframe length to the multi-pulse generation parameter setup circuit 33.

[0084] The CELP coding network 72 encodes an input signal according to the frame length set up by the coding parameter-control circuit 71, subframe length and bit distribution and the amount of multi-pulse-coding permissible operations, and pulse-number P set up in the multi-pulse generation parameter setup circuit 33, each number of pulse

candidate positions M (p) and its candidate position.

[0085] Drawing 10 is also referred to and operation of the CELP coding network 72 is explained.

[0086] Only operation of a multi-pulse search circuit differs compared with the CELP coding equipment which explained this CELP coding network 72 by drawing 5. Therefore, only operation of the multi-pulse search circuit 501 is explained here.

[0087] The multi-pulse search circuit 501 performs the same operation as the multi-pulse search circuit 401 explained by drawing 5, and chooses the multi-pulse signal $C_j(n)$ which minimizes several 4. In this case, preliminary selection is performed so that the amount of operations spent on coding of a multi-pulse signal may not exceed the amount of multi-pulse-coding permissible operations inputted through the input terminal 218. Moreover, the multi-pulse search circuit 501 outputs the index j which corresponds while outputting the selected multi-pulse signal $C_j(n)$ and its regenerative signal $S_{Cj}(n)$ to the gain search circuit 211 to a multiplexer 13 through an output terminal 216.

[0088]

[Effect of the Invention] As explained above, it is effective in not only a bit rate but a shell, and the coding delay and the amount of operations which were generated based on the bit rate and coding delay which the frame length which is a coding batch was made [delay] adjustable, and had the parameter required for coding of a multi-pulse signal specified being controllable by this invention. Therefore, since according to this invention it can respond with the same coding decode equipment to make a bit rate as fewer [than coding delay of voice mail etc.] as possible when you want to shorten coding delay as much as possible by the video conference system etc. or, the scale of coding decode equipment can be made small.

[Translation done.]

* NOTICES *

Japan Patent Office is not responsible for any
damages caused by the use of this translation.

- 1.This document has been translated by computer. So the translation may not reflect the original precisely.
- 2.**** shows the word which can not be translated.
- 3.In the drawings, any words are not translated.

DESCRIPTION OF DRAWINGS

[Brief Description of the Drawings]

[Drawing 1] It is the block diagram showing the 1st example of the voice coding decode equipment by this invention.

[Drawing 2] It is a block diagram for explaining the CELP coding network shown in drawing 1.

[Drawing 3] It is a block diagram for explaining the CELP decoder circuit shown in drawing 1.

[Drawing 4] It is the block diagram showing the 2nd example of the voice coding decode equipment by this invention.

[Drawing 5] It is a block diagram for explaining the CELP coding network shown in drawing 4.

[Drawing 6] It is a block diagram for explaining the CELP decoder circuit shown in drawing 4.

[Drawing 7] It is the block diagram showing the 3rd example of the voice to digital converter by this invention.

[Drawing 8] It is a block diagram for explaining the CELP coding network shown in drawing 7.

[Drawing 9] It is the block diagram showing the 4th example of the voice to digital converter by this invention.

[Drawing 10] It is a block diagram for explaining the CELP coding network shown in drawing 9.

[Drawing 11] It is the block diagram showing an example of conventional voice coding decode equipment.

[Description of Notations]

11, 15, 31, 34 Coding parameter-control circuit

12 32 CELP coding network

16 35 CELP decoder circuit

33 36 Multi-pulse-coding parameter setup circuit

[Translation done.]

* NOTICES *

Japan Patent Office is not responsible for any damages caused by the use of this translation.

1. This document has been translated by computer. So the translation may not reflect the original precisely.
2. **** shows the word which can not be translated.
3. In the drawings, any words are not translated.

CLAIMS

[Claim(s)]

[Claim 1] The excitation signal of an input sound signal is expressed with two or more pulses. with the aforementioned excitation signal It is the voice to digital converter which has a voice coding means to determine that the aforementioned excitation signal will minimize distortion between the reproduction sound signals and the aforementioned input sound signals which excite the linear-prediction composition filter specified by the linear predictor coefficients of the aforementioned input sound signal, and are obtained. It is the voice to digital converter which is equipped with the control circuit which generates a control parameter according to the specified control information, and is characterized by the aforementioned voice coding means encoding the aforementioned input sound signal according to the aforementioned control parameter.

[Claim 2] The excitation signal of an input sound signal is expressed with two or more pulses. with the aforementioned excitation signal It is the voice to digital converter which has a voice coding means to determine that the aforementioned excitation signal will minimize distortion between the reproduction sound signals and the aforementioned input sound signals which excite the linear-prediction composition filter specified by the linear predictor coefficients of the aforementioned input sound signal, and are obtained. It is the voice to digital converter which is equipped with the control circuit which receives the bit rate and coding delay which were specified as control information, and generates a control parameter according to this control information, and is characterized by the aforementioned voice coding means encoding the aforementioned input sound signal according to the aforementioned control parameter.

[Claim 3] The excitation signal of an input sound signal is expressed with the multi-pulse signal which consists of two or more pulses. with the aforementioned excitation signal It is the voice to digital converter which has a voice coding means to determine that the aforementioned excitation signal will minimize distortion between the reproduction sound signals and the aforementioned input sound signals which excite the linear-prediction composition filter specified by the linear predictor coefficients of the aforementioned input sound signal, and are obtained. The control circuit which receives the bit rate and coding delay which were specified as control information, and generates a control parameter according to this control information, It has the setting circuit which sets up a parameter required for coding of the aforementioned multi-pulse signal as an active parameter according to a **** parameter. the parameter beforehand defined among the aforementioned control parameters gives -- having -- eye this ** -- laws -- The aforementioned voice coding means is a voice to digital converter characterized by encoding the aforementioned input sound signal based on the aforementioned control parameter and the aforementioned active parameter.

[Claim 4] The excitation signal of an input sound signal is expressed with two or more pulses. with the aforementioned excitation signal The bit rate which is the voice to digital converter which has a voice coding means to determine that the aforementioned excitation signal will minimize distortion between the reproduction sound signals and the aforementioned input sound signals which excite the linear-prediction composition filter specified by the linear predictor coefficients of the aforementioned input sound signal, and are obtained, and was specified, It is the voice to digital converter which is equipped with coding delay and the control circuit which receives the amount of operations as control information, and generates a control parameter according to this control information, and is characterized by the aforementioned voice coding means encoding the aforementioned input sound signal according to the aforementioned control parameter.

[Claim 5] The excitation signal of an input sound signal is expressed with the multi-pulse signal which consists of two or more pulses. with the aforementioned excitation signal The bit rate which is the voice to digital converter which has a voice coding means to determine that the aforementioned excitation signal will minimize distortion between the reproduction sound signals and the aforementioned input sound signals which excite the linear-prediction composition filter specified by the linear predictor coefficients of the aforementioned input sound signal, and are obtained, and was specified, Coding delay and the control circuit which receives the amount of operations as control information, and generates a control parameter according to this control information, It has the setting circuit which sets up a parameter required for coding of the aforementioned multi-pulse signal as an active parameter according to a **** parameter. the parameter beforehand defined among the aforementioned control parameters gives -- having -- eye this ** -- laws -- The aforementioned voice coding means is a voice to digital converter characterized by encoding the aforementioned input sound signal based on the aforementioned control parameter and the aforementioned active parameter.

[Claim 6] Voice decode equipment which is characterized by providing the following and which receives coding voice data and reproduces a reproduction sound signal from the aforementioned coding voice data. The aforementioned coding voice data is a control circuit which includes control information while having the excitation signal and linear-prediction

composition filter factor of a sound signal, and generates a control parameter according to the aforementioned control information. A decode means to excite the linear-prediction composition filter which decodes the aforementioned excitation signal and the aforementioned linear-prediction composition filter factor from the aforementioned coding voice data according to the aforementioned control parameter, and is specified by the aforementioned linear-prediction composition filter factor according to the aforementioned excitation signal, and to reproduce the aforementioned reproduction sound signal.

[Claim 7] Voice coding decode equipment characterized by having voice decode equipment indicated by the voice to digital converter indicated by the claim 1 and the claim 6.

[Claim 8] Voice decode equipment which is characterized by providing the following and which receives coding voice data and reproduces a reproduction sound signal from the aforementioned coding voice data. The aforementioned coding voice data is a control circuit which includes a bit rate and coding delay while having the excitation signal and linear-prediction composition filter factor of a sound signal, and generates a control parameter based on the aforementioned bit rate and the aforementioned coding delay. A decode means to excite the linear-prediction composition filter which decodes the aforementioned excitation signal and the aforementioned linear-prediction composition filter factor from the aforementioned coding voice data according to the aforementioned control parameter, and is specified by the aforementioned linear-prediction composition filter factor according to the aforementioned excitation signal, and to reproduce the aforementioned reproduction sound signal.

[Claim 9] Voice coding decode equipment characterized by having voice decode equipment indicated by the voice to digital converter indicated by the claim 2 or the claim 4 and the claim 8.

[Claim 10] Voice decode equipment which is characterized by providing the following and which receives coding voice data and reproduces a reproduction sound signal from the aforementioned coding voice data. The aforementioned coding voice data is a control circuit which the aforementioned excitation signal is expressed with the multi-pulse signal which consists of two or more pulses, and generates a control parameter based on the aforementioned bit rate and the aforementioned coding delay including a bit rate and coding delay while having the excitation signal and linear-prediction composition filter factor of a sound signal. the parameter beforehand defined among the aforementioned control parameters gives -- having -- eye this ** -- laws -- the setting circuit which sets up a parameter required for coding of the aforementioned multi-pulse signal as an active parameter according to a **** parameter A decode means to excite the linear-prediction composition filter which decodes the aforementioned excitation signal and the aforementioned linear-prediction composition filter factor from the aforementioned coding voice data based on the aforementioned control parameter and the aforementioned active parameter, and is specified by the aforementioned linear-prediction composition filter factor according to the aforementioned excitation signal, and to reproduce the aforementioned reproduction sound signal.

[Claim 11] Voice coding decode equipment characterized by having voice decode equipment indicated by the voice to digital converter indicated by the claim 3 or the claim 5 and the claim 10.

[Translation done.]

***This Page is Inserted by IFW Indexing and Scanning
Operations and is not part of the Official Record**

BEST AVAILABLE IMAGES

Defective images within this document are accurate representations of the original documents submitted by the applicant.

Defects in the images include but are not limited to the items checked:

- BLACK BORDERS**
- IMAGE CUT OFF AT TOP, BOTTOM OR SIDES**
- FADED TEXT OR DRAWING**
- BLURRED OR ILLEGIBLE TEXT OR DRAWING**
- SKEWED/SLANTED IMAGES**
- COLOR OR BLACK AND WHITE PHOTOGRAPHS**
- GRAY SCALE DOCUMENTS**
- LINES OR MARKS ON ORIGINAL DOCUMENT**
- REFERENCE(S) OR EXHIBIT(S) SUBMITTED ARE POOR QUALITY**
- OTHER: _____**

IMAGES ARE BEST AVAILABLE COPY.

As rescanning these documents will not correct the image problems checked, please do not report these problems to the IFW Image Problem Mailbox.